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Patentanmeldung Nr. Patent application No. Demande de brevet n°

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Der Präsident des Europäischen Patentamts;
Im Auftrag

For the President of the European Patent Office

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Blatt 2 der Bescheinigung
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IMPROVING MODELING OF AUDIO SIGNALS BY MODIFYING TRANSIENT LOCATIONS

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We propose a method for obtaining an improved representation of transients in audio signals. The representation is based on a damped sinusoidal model. To improve the representation, transient locations are modified in such a way that a transient can start only at the beginning of a sinusoidal segment. The introduced modifications facilitate a reduction of the number of damped sinusoids needed to model a transient well and the elimination of pre-echo artifacts. With a listening test we verify that the modifications do not result in a perceptual difference between the original and modified audio signals.

1. INTRODUCTION

Parametric coding of audio is a popular tool for representing audio signals at very low bit rates [1, 2, 3, 4, 5]. In a parametric audio coder, a signal is represented by a model, and parameters of the model are estimated and encoded. A popular parametric representation of audio signals is based on a decomposition of an original signal into three components: a transient component, a tonal (sinusoidal) component, and a noise component (e.g., [1, 4, 5]). Having a dedicated model for the transient component proved to be beneficial for parts of audio signals with sharp attacks, because sinusoidal and noise models cannot represent those perceptually important events efficiently [6].

We propose a method for an improved representation of transients. It was shown in [7] that transients can be modeled efficiently using sinusoids with exponentially-modulated amplitudes (damped sinusoids). An audio signal is analyzed on a segment-by-segment basis, and each segment is represented as a sum of damped sinusoids. A problem occurs when a transient does not start at the beginning of a segment. Compared to the case where a transient starts at the beginning of a segment, the number of damped sinusoids needed to model the transient well increases considerably. If a transient is not modeled properly, the modeling error is distributed over the whole segment, resulting in audible pre-echoes. Different methods have been used to solve this problem:

- Allow a one-sample-precision (full-precision) variable segmentation of the signal, such that transients will always start at the beginnings of segments (e.g., [1]).
- Allow a switching between a long and a short window defining analysis segments, such that short windows are used for parts of an audio signal with sharp attacks (e.g., MPEG-1 Layer III audio coding algorithm [8]). In this case, the segmentation is defined simply by the lengths of the long and the short windows.

In this paper, we use a restricted time segmentation. By restricted segmentation we mean that the segment lengths are defined by integer multiples of a predefined minimum segment length, say 5 ms. Given such a restricted time segmentation, we modify the transient component of the audio signal such that a transient can start only at the beginning of a segment. This will result in an efficient representation of transients with damped sinusoids. The advantages of this method as compared to the full-precision variable segmentation are the following:

- The restricted segmentation significantly simplifies the analysis procedure in an audio coder.
- The restricted segmentation results in a reduction of the number of bits needed to describe the segmentation.

The remainder of this paper is organized as follows. The procedure to modify transient locations is described in Section 2. Modeling with damped sinusoids is described in Section 3. Results of computer simulations and listening tests are presented in Section 4. Finally, conclusions are summarized in Section 5.

2. MODIFICATION OF TRANSIENT LOCATIONS

In [9] we presented a method for modifying transient locations in an audio signal. The transient component of the audio signal is estimated using a model based on duality between the time and the frequency domain, as presented in [10]. This transient model is good for very short transients, i.e., with a sharp attack and a fast decay. Transient locations are modified by modifying parameters of a frequency-domain representation of the transient component.

This paper presents an improved method for modifying transient locations. In this new method, an audio signal is modified in the following steps:

1. The beginnings and ends of transients are detected using an energy-based approach with two sliding rectangular windows, as presented in [11].
2. The samples between the beginning and the end of each transient are shifted (essentially, cut-and-paste) to the locations specified by sinusoidal segmentation.
3. The signal parts in between transients are time-warped in order to fill the intervals between the shifted transients.

The advantages of the new transient modification method over the one presented in [9] are the following:

- The transient detection model of [11] provides good results also for transients with slow decay.
- The time-warping of the signal parts in between transients is based on the knowledge of properties of sound perception, such as pitch perception and temporal masking effects.
- The new modification method results in a lower computational complexity.

The transient detection approach of [11] used in step 1 is based on the evaluation of the criterion function, $C(n)$:

$$C(n) = \log\left(\frac{E_R(n)}{E_L(n)}\right) \cdot E_R(n), \qquad (1)$$

$$E_L(n) = \sum_{k=n-N}^{n-1} s^2(k), \qquad E_R(n) = \sum_{k=n+1}^{n+N} s^2(k),$$

where $E_L(n)$ and $E_R(n)$ are the energies of the input signal s within length-N rectangular windows on the left- and right-hand side of a time sample n. Significant peaks of the criterion function $C(n)$ correspond to the starts of transients.

Step 2 of the new transient modification method is obvious. We now describe step 3 of the modification method. Due to modification of transient locations, the distance between two transients can become longer or shorter. In order to fill the interval between the shifted transients, the signal part in between has to be time-warped correspondingly. The time-warping of the signal is done in such a way that it preserves the correct amplitudes of the edge points of the signal part in between the transients. Thus, no discontinuities are introduced just before or after a transient. The signal in between transients is stretched or compressed in time. To compute the amplitudes at the new integer sampling instances based on the known amplitudes of the original samples an approximation of the ideal bandlimited interpolation based on *sinc* functions is used. To compute the amplitude of each new sample, amplitudes of eight original samples are used, four at each side of the new sample. A hanning window is used to limit the length of the *sinc* functions.

For tonal signals, a stretching or compressing the signal in time results in a corresponding change of fundamental frequency, f_0. The goal of the modification procedure is to ensure that the induced modification of f_0 is not audible. Therefore, the following algorithm is proposed for time-warping a signal part in between two shifted transients (the steps are illustrated in Figure 1 for the case where the length of the signal between two shifted transients is longer than the original; the opposite case is treated similarly):

a) If the required change in length of a signal part in between two transients results in the change of f_0 by no more than 0.2 %, then simply use the time-warping method as described above (Figure 1a). Else go to step b.

Motivation: from the literature on psychoacoustics it is known that changing f_0 of a tonal sound by 0.2 % can be audible [12]. Our experiments verified this result.

b) Split the signal part in between two transients into two nonoverlapping intervals: the first interval is located directly after the end of the first transient and lasts 10 ms (interval 1 in Figure 1b), and the second interval is the remaining part, i.e. it lasts until the beginning of the second transient (interval 2 in Figure 1b). The lengths of the two intervals are modified by a different amount. If the required change in length of the signal part in between two transients can be done by changing f_0 in the interval 1 by no more than 2 % and in the interval 2 by no more than 0.2 %, then time-warp the signal in the two intervals correspondingly. Else go to step c.

Motivation: the interval directly after the end of a transient is characterized by a strong masking effect from the transient. Therefore, larger changes of the signal in this interval are possible before they become audible. Our experiments verified that a change of f_0 by no more than 2 % in the 10 ms interval directly after the end of a transient is inaudible.

c) Time-warp the signal in the two intervals such that the

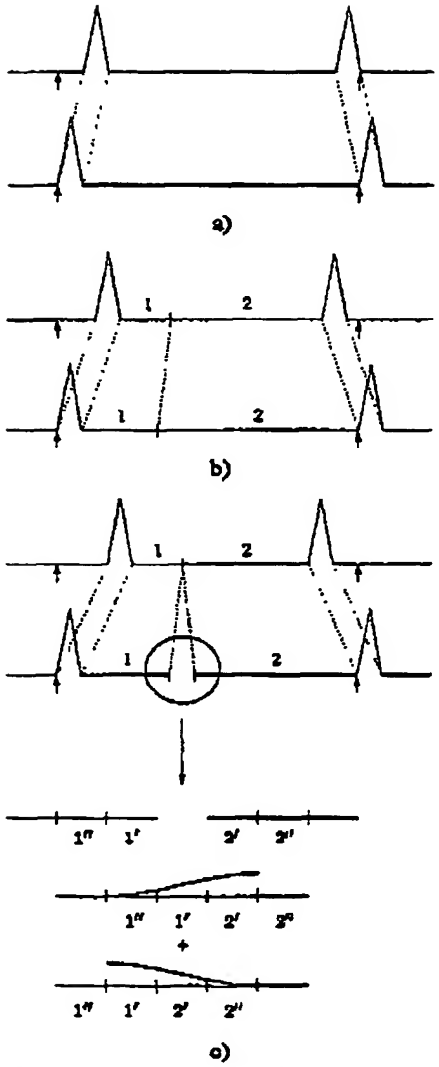

Figure 1: Modification of transient locations. The new locations of transient beginnings are depicted with small arrows. The signal part in between two transients becomes longer. Steps a, b, c are explained in Section 2.

sulting change of f_0 is no more than 2 % in the interval 1 and no more than 0.2 % in the interval 2. If the resulting change in length is not sufficient to fill the distance between the shifted transients then apply an overlap-add procedure with a hanning window using samples from the two intervals in order to increase or decrease the length of the signal. To ensure a smooth transition between two intervals, the length of the overlap-add region is chosen to be larger than required to obtain a correct length of the signal in between two transients (Figure 1c).

3. MODELING WITH DAMPED SINUSOIDS

It was shown in [7] that a transient can be modeled efficiently using a damped sinusoidal model. This model aims at approximating a signal s by a sum of, say M, sinusoids with exponentially modulated amplitudes, i.e.,

$$\tilde{s}(n) = \sum_{m=1}^{M} a_m e^{d_m n} \cos(\omega_m n + \varphi_m), \quad n = 0, \ldots, N-1, \quad (2)$$

where $a_m, d_m, \omega_m, \varphi_m \in \mathbb{R}$ denote the amplitude, damping coefficient[1], angular frequency and phase of the mth sinusoidal component, respectively. $N \in \mathbb{N}$ is the segment length.

The sinusoidal parameters a_m, d_m, ω_m and φ_m can be selected with a number of methods, including spectral peak-picking, subspace-based analysis techniques and analysis-by-synthesis methods. For the experiments described in this paper we used the matching pursuit algorithm [13], a particular analysis-by-synthesis method. The matching pursuit algorithm is a greedy iterative algorithm which projects at each iteration a signal onto the function (in our case a damped sinusoid) that best matches the signal and subtracts this projection to form a residual signal to be approximated in the next iteration.

In order to find an "optimal" time segmentation we used the algorithm proposed in [14]. By optimal we mean optimal in a rate-distortion sense. This algorithm divides the input signal s into non-overlapping segments and tries, by combining these segments, to find the partitioning of s that minimizes the distortion given a target bit budget or a given number of sinusoidal components. Under the assumption of additivity of rate and distortion over the constituent segments, the global optimal segmentation is found by first minimizing the rate vs distortion for each segment independently, and then, using dynamic programming, find the optimal segmentation by combining these optimal encoded segments. By doing so, the algorithm gives the optimal time segmentation of s, as well as the number of sinusoidal components to allocate to the individual segments.

4. EXPERIMENTAL RESULTS

Below, we present results of computer simulations and listening tests with audio signals. The signals are mono, sampled at 44.1 kHz. The test excerpts include castanets, bass, ABBA, Celine Dion, Metallica, harpsichord, Suzanne Vega. Transient locations are modified according to time grid of 220 samples (ca 5 ms).

[1]The damping coefficient d_m can be any real number. Positive values of d_m, therefore, correspond to expanding amplitudes rather than to truly damped amplitudes.

Excerpt	Duration, s	# detected transients	Correct responses, %
castanets	7.1	43	57.5
bass	10.8	16	52.5
ABBA	9.9	29	45.0
Celine Dion	12.8	26	52.5
Metallica	10.1	19	52.5
harpsichord	11.7	9	40.0
Suzanne Vega	10.1	13	42.5

Table 1: Results of the listening test on audibility of signal modifications which include shifting transients and time-warping the signal parts in between transients.

It is important to verify that the introduced modifications do not result in any audible difference between the original and modified audio signals. To do that we performed a subjective listening test in which signal triplets AOB were presented to listeners. Here O is the original signal, A or B is the original signal and B or A is the modified signal. The task of a listener was to respond whether the modified signal was A or B. For each test excerpt, the triplets AOB were presented to a listener 5 times, each time the position of the modified signal (A or B) was changed randomly. Eight experienced listeners participated in the test. The results averaged over all listeners are presented in Table 1. They confirm that the introduced modifications are not audible.

Next, we illustrate the improvement due to the modification procedure. We study the performance of a damped sinusoidal model for an original signal (i.e., transients start at arbitrary locations) and for a modified signal (i.e., transients can start only at the beginnings of sinusoidal segments). The methods used to evaluate the performance are the same as in [9]. The performance is studied in terms of signal-to-noise ratio (SNR) versus the number of damped sinusoids and is well illustrated in Figure 2, where it is presented for a particular transient of the castanets signal. It is evident that more sinusoids are needed to model the transient with a certain quality in the case when the transient does not start at the beginning of a sinusoidal segment. The lower plots f Figures 3 and 4 show the reconstruction with 25 damped sinusoids of the original and the modified transients, respectively. The original transient does not start at the beginning of the segment and, as a result, the modeling error is distributed to samples before the transient. This results in a clearly audible pre-echo. On the other hand, the modified transient starts at the beginning of the segment and, as a result, the pre-echo problem is eliminated.

5. CONCLUSIONS

In this paper, we elaborated on the idea of modifying transient locations in an audio signals for improved modeling and coding of audio. We presented a new method for modifying transient locations. The introduced modifications facilitate an efficient representation of transients with damped sinusoids and the elimination of pre-echo artifacts. We also verified that the modifications are not audible.

It has to be noted, however, that a straightforward application of the modification procedure is not suitable for stereo signals. The reason for this is that an independent modification of transient locations in two channels may destroy the original stereo image. We are currently working on this issue.

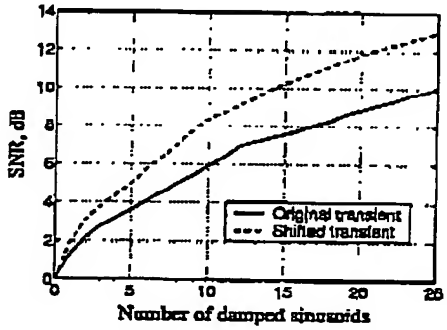

Figure 2: Performance of a damped sinusoidal model in the case of a restricted segmentation for an original and a shifted transient. The minimum segment length is 5 ms.

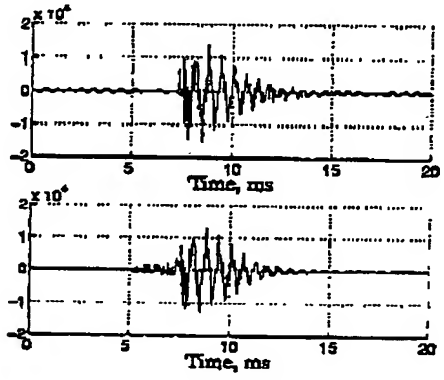

Figure 3: The original transient and its reconstruction with 25 damped sinusoids. The minimum segment length is 5 ms.

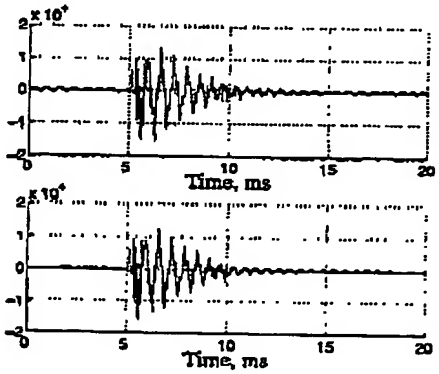

Figure 4: The shifted transient and its reconstruction with 25 damped sinusoids. The minimum segment length is 5 ms.

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CLAIMS:

1. A method of encoding a signal to obtain an encoded signal, the method comprising the steps of:
estimating a transient in the signal,
modifying a location of the estimated transient to obtain a modified transient having a location closer to a grid location than the estimated transient location, the grid location being specified by a predefined time grid, and
including a representation of the modified transient in the encoded signal.

2. An encoder for encoding a signal to obtain an encoded signal, the encoder comprising:
means for estimating a transient in the signal,
means for modifying a location of the estimated transient to obtain a modified transient having a location closer to a grid location than the estimated transient location, the grid location being specified by a predefined time grid, and
means for including a representation of the modified transient in the encoded signal.

3. An encoded signal comprising representations of modified transients having a location approximating a grid location, the grid location being specified by a predefined time grid.

4. A storage medium on which a signal as claimed in claim 3 has been stored.

Fig. 5

Fig. 5 shows an audio encoder according to an embodiment of the invention. This encoder estimates a transient in the audio signal, modifies a location of the estimated transient to obtain a modified transient having a location closer to a grid location than the estimated transient location, the grid location being specified by a predefined time grid, and includes a representation of the modified transient in the encoded audio signal. The audio encoder of Fig. 5 may be included in a transmitter, which transmitter further comprises a source for obtaining the audio signal and an output unit for outputting the encoded audio signal. The audio signal may include speech.